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APPLICATION NO.	FILING DATE	FIRST NAMED INVENTOR	ATTORNEY DOCKET NO.	CONFIRMATION NO.
09/346,884	07/02/1999	NIRAT BHUPESH SHAH	14013-23	3005

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EXAMINER	
LY, ANH VU H	
ART UNIT	**PAPER NUMBER**
2667	22

DATE MAILED: 07/23/2004

Please find below and/or attached an Office communication concerning this application or proceeding.

Office Action Summary	Application No.	Applicant(s)
	09/346,884	SHAH, NIRAT BHUPESH
	Examiner	**Art Unit**
	Anh-Vu H Ly	2667

-- The MAILING DATE of this communication appears on the cover sheet with the correspondence address --

Period for Reply

A SHORTENED STATUTORY PERIOD FOR REPLY IS SET TO EXPIRE 3 MONTH(S) FROM THE MAILING DATE OF THIS COMMUNICATION.

- Extensions of time may be available under the provisions of 37 CFR 1.136(a). In no event, however, may a reply be timely filed after SIX (6) MONTHS from the mailing date of this communication.
- If the period for reply specified above is less than thirty (30) days, a reply within the statutory minimum of thirty (30) days will be considered timely.
- If NO period for reply is specified above, the maximum statutory period will apply and will expire SIX (6) MONTHS from the mailing date of this communication.
- Failure to reply within the set or extended period for reply will, by statute, cause the application to become ABANDONED (35 U.S.C. § 133). Any reply received by the Office later than three months after the mailing date of this communication, even if timely filed, may reduce any earned patent term adjustment. See 37 CFR 1.704(b).

Status

1)☒ Responsive to communication(s) filed on *01 June 2004*.

2a)☐ This action is **FINAL**. 2b)☒ This action is non-final.

3)☐ Since this application is in condition for allowance except for formal matters, prosecution as to the merits is closed in accordance with the practice under *Ex parte Quayle*, 1935 C.D. 11, 453 O.G. 213.

Disposition of Claims

4)☒ Claim(s) *1-20* is/are pending in the application.

4a) Of the above claim(s) _____ is/are withdrawn from consideration.

5)☐ Claim(s) _____ is/are allowed.

6)☒ Claim(s) *1-20* is/are rejected.

7)☐ Claim(s) _____ is/are objected to.

8)☐ Claim(s) _____ are subject to restriction and/or election requirement.

Application Papers

9)☐ The specification is objected to by the Examiner.

10)☐ The drawing(s) filed on _____ is/are: a)☐ accepted or b)☐ objected to by the Examiner.

Applicant may not request that any objection to the drawing(s) be held in abeyance. See 37 CFR 1.85(a).

Replacement drawing sheet(s) including the correction is required if the drawing(s) is objected to. See 37 CFR 1.121(d).

11)☐ The oath or declaration is objected to by the Examiner. Note the attached Office Action or form PTO-152.

Priority under 35 U.S.C. § 119

12)☐ Acknowledgment is made of a claim for foreign priority under 35 U.S.C. § 119(a)-(d) or (f).

a)☐ All b)☐ Some * c)☐ None of:

1.☐ Certified copies of the priority documents have been received.

2.☐ Certified copies of the priority documents have been received in Application No. _____.

3.☐ Copies of the certified copies of the priority documents have been received in this National Stage application from the International Bureau (PCT Rule 17.2(a)).

* See the attached detailed Office action for a list of the certified copies not received.

Attachment(s)

1) ☒ Notice of References Cited (PTO-892)

2) ☐ Notice of Draftsperson's Patent Drawing Review (PTO-948)

3) ☐ Information Disclosure Statement(s) (PTO-1449 or PTO/SB/08) Paper No(s)/Mail Date _____.

4) ☐ Interview Summary (PTO-413) Paper No(s)/Mail Date. _____ .

5) ☐ Notice of Informal Patent Application (PTO-152)

6) ☐ Other: _____.

DETAILED ACTION

Continued Examination Under 37 CFR 1.114

1. A request for continued examination under 37 CFR 1.114, including the fee set forth in 37 CFR 1.17(e), was filed in this application after final rejection. Since this application is eligible for continued examination under 37 CFR 1.114, and the fee set forth in 37 CFR 1.17(e) has been timely paid, the finality of the previous Office action has been withdrawn pursuant to 37 CFR 1.114. Applicant's submission filed on June 01, 2004 has been entered.

Claim Objections

2. Claim 17 is objected to because of the following informalities: in line 22 "devices, _upon further" is unclear. Appropriate correction is required.

Claim Rejections - 35 USC § 103

The following is a quotation of 35 U.S.C. 103(a) which forms the basis for all obviousness rejections set forth in this Office action:

> (a) A patent may not be obtained though the invention is not identically disclosed or described as set forth in section 102 of this title, if the differences between the subject matter sought to be patented and the prior art are such that the subject matter as a whole would have been obvious at the time the invention was made to a person having ordinary skill in the art to which said subject matter pertains. Patentability shall not be negatived by the manner in which the invention was made.

3. Claims 1-13 and 17-19 are rejected under 35 U.S.C. 103(a) as being unpatentable over Vargo et al (US Patent No. 6,356,545) in view of Bauer et al (US Pub No. 2001/0008556 A1) and further in view of Riddle (US Patent No. 6,175,856). Hereinafter, referred to as Vargo, Bauer, and Riddle.

With respect to claims 1 and 17, the limitations recited in claim 1, a DSP module responsive to an analog signal from one of the telephone devices and operative to convert analog

telephone signal to digital telephone signal and further operative to packetize digital telephone signal for transmission to a remotely-located router device, are inherent to Vargo.

Vargo discloses in Fig. 1, an operation of the Internet telephone system (communication system). Wherein, a call is initiated in North America (first telephone device) over a PSTN gateway server 10a (router device) from a PSTN 11a over the Internet 17 (packet switching network) to Japan and Taiwan (second telephone device).

Further, according to Fig. 1, for a call to take place over the Internet (packet switching network), the received analog signal (analog telephone signal), from the call initiator (first telephone device) to the call receiver in Japan (second telephone device), must be digitized into packets (converted analog signal to digital signal) and transferred to the router (remotely-located router device) for routing packets through the packet network. Besides, all involved processes stated above are well known in the art in VoIP transmissions.

The limitation "the router device and the remotely-located device initially mutually negotiating to utilize a first type of codec" is inherent to Vargo. Vargo discloses (col. 10, lines 46-67 and Fig. 11a) assuming the voice port begins with the commercially available TrueSpeech codec algorithm (first type of codec), which encodes speech at 8.5kbits/sec and with no redundancy. This means that both parties must initially negotiated for using the same codec to understand one another.

After noticing dropped packets (detection of degradation in the quality of the voice information), the voice port (DSP module for renegotiated the use of a second type of codec) adjusts by selecting (switching codec type) the Voxware 2.9kbits/sec algorithm (second type of

codec) having somewhat lower sound quality (while conversation is taking place), but with two level redundancy error correction.

Vargo discloses (col. 2, lines 51-54) that the architecture enables a dynamic change of codec from packet to packet in the same voice data stream to adapt to changing network conditions. Vargo does not disclose that wherein, during communications between the remotely-located router device and the DSP module, the type of coded being utilized is repeatedly, mutually, renegotiated to dynamically change compression techniques to adjust for network usage thereby optimizing the use of network capacity and throughput. Bauer discloses (col. 3, paragraph 27 to paragraph 30) that the initiating device inserts a notification in a field of the packet header to inform the recipient device that subsequent packets will be encoded with a different specified algorithm, until further notice. Thereafter, the recipient device can load the appropriate coded to properly decode the received packets. In a further variation, the notification of a coded change or the current code can be repeatedly included in the packet header at periodic intervals, or repeated a predetermined number of times in successive packets.

Therefore, it would have been obvious to one having ordinary skill in the art at the time the invention was made to include the features of having the source and the destination, repeatedly, mutually, renegotiated for the new type of codec in response to the network conditions in Vargo's system, as suggested by Bauer, to accommodate QoS and actively manage the bandwidth of a packet telephony system.

Vargo does not disclose that each sending to the other a list of one or more codecs that each supports and each deciding to use a mutually supported codec through the use of a predetermined protocol. Riddle discloses in Figs. 5 and 6, that the sender/initiator (router

device) and the receiver (remotely located router device as considered by the examiner) exchanging information regarding list of codecs that each can support and selecting a best codec from the list of exchanged codecs (each sending to the other a list of one or more codecs that each supports and each deciding to use a mutually supported codec). Riddle discloses in Figs. 1 and 2, a system for supporting teleconference between plurality of workstations and routers connecting different networks. Such system is implemented by a specific protocol therefore the step of exchanging information is also carried out by using such specific protocol (use of a predetermined protocol).

It would have been obvious to one having ordinary skill in the art at the time the invention was made to include the features of exchanging list of codecs between the sender and the receiver and selecting a best codec from the lists of exchanged codecs in Vargo's system, as suggested by Riddle, to reduce time in codec negotiations in a system.

With respect to claims 2-4, Vargo disclose (col. 4, lines 48-51) that the teleport is designed to be able to switch codecs between one data packet and the next in the same data stream. Vargo does not disclose that wherein switching between the codes is initiated by a user of one of the telephone devices and wherein a predetermined code is assigned to each codec, the user specifies the type of codec to be switched to by entering the predetermined code corresponding to a desired codec into one of user telephone devices and predetermined code is programmably-alterable. However, switching initiated by a user and predetermined code are well known in the art such a TV remote controller, wherein a user can select different channels to view and wherein the remote controller can be programmed to store a number of channels with

associated "hot keys". Wherein, each "hot key" is corresponded with a channel and a user can press that "hot key" to turn to that specific channel. User can re-program the remote controller to different "hot keys" associated with different channels at another time. It would have been obvious to one having ordinary skill in the art at the time the invention was made to include a method of user initiating and assigned predetermined code, which is re-programmable, for each codec in Vargo's system, to increase system's functionalities.

With respect to claim 5, Vargo discloses (col. 11, lines 18-19) that voice port 61 responds to changing network conditions (detecting lower bandwidth available on the packet switching network) to maintain speech quality. Further Vargo discloses (col. 11, lines, 20-22) that it is possible to vary the size of the individual packets or to vary the bundling-of the packets (switching from a codec resulting in the use of larger packet sizes to a codec resulting in smaller packet sizes) by techniques that are well known in the art.

With respect to claims 6 and 8, the limitation "wherein the router device automatically detecting the lower and higher bandwidth" is addressed in the rejection of claim 5. Wherein, Vargo discloses that voice port 61 responds to changing network conditions to maintain speech quality.

With respect to claim 7, Vargo discloses (col. 11, lines 18-19) that voice port 61 responds to changing network conditions (detecting higher bandwidth available on the packet switching network) to maintain speech quality. Further Vargo discloses (col. 11, lines, 20-22) that it is

possible to vary the size of the individual packets or to vary the bundling-of the packets (switching from a codec resulting in the use of smaller packet sizes to a codec resulting in higher packet sizes) by techniques that are well known in the art.

With respect to claim 9, Vargo discloses (col. 10, lines 46-67 and Fig. 11a) that after noticing dropped packets (remotely-located router device detects the degradation in the quality of the voice information), the voice port adjusts by selecting the Voxware 2.9kbits/sec algorithm having somewhat lower sound quality, but with two level redundancy error correction.

With respect to claim 10, the limitation recited in claim 10 is addressed in the rejection of parent claim 1. Wherein, Vargo discloses that after noticing dropped packets (loss of one or more packets), the voice port adjusts by selecting the Voxware 2.9kbits/sec algorithm having somewhat lower sound quality, but with two level redundancy error correction.

With respect to claim 11, the limitation recited in claim 11 is addressed in the rejection of claim 10. Wherein Vargo discloses that after noticing dropped packets (threshold defines the number of lost packets), the voice port adjusts by selecting the Voxware 2.9kbits/sec algorithm having somewhat lower sound quality, but with two level redundancy error correction.

With respect to claim 12, the limitation "wherein a plurality of packets form a message and each packet includes a sequence number indicative of the position of the packet with respect to other packets in the same message, the sequence numbers of the same message being in

sequential order" is addressed in the rejection of parent claim 1. Wherein Vargo discloses that a stream of voice data 200 includes a plurality of data packets numbered 1 through 10, where each packet further contains a plurality of data bytes indicated by the letters in Fig. 8(a) to 8(d).

The limitation, wherein a loss of packets is detected when one or more sequence numbers are missing from the received packets of the same message, is addressed in rejection of claim 11. Wherein, Vargo discloses that after noticing dropped packets, the voice port adjusts by selecting the Voxware 2.9kbits/sec algorithm having somewhat lower sound quality, but with two level redundancy error correction.

With respect to claim 13, Vargo discloses (col. 1, lines 40-43) that since Internet is built to transfer data packets rather than continuous streams of sound, there may be delays and losses. Further, Vargo discloses that voice port 61 responds to changing network conditions (degradation in the quality of the voice information is due to an intolerable delay associated with the time for a packet to travel between the router device and the remotely-located router device) to maintain speech quality.

With respect to claim 18, "wherein codec negotiation is performed pursuant to H.245 protocol" is inherent to Vargo. Vargo discloses Internet telephone systems; wherein, H.245 protocol is known for exchanging signaling messages.

With respect to claim 19, Vargo discloses (col. 10, lines 46-67 and Fig. 11a) assuming the voice port begins with the commercially available TrueSpeech codec algorithm, which encodes speech at 8.5kbits/sec (first type of codec utilizes a compression/decompression

algorithm defined by any one of the standards: G.711, G.726, G.729, or G723.1) and with no redundancy.

After noticing dropped packets, the voice port adjusts by selecting the Voxware 2.9kbits/sec algorithm (second type of codec utilizes a compression/decompression algorithm defined by any one of the standards: G.711, G.726, G.729, or G723.1) having somewhat lower sound quality but with two level redundancy error correction.

4. Claims 14-16 and 20 are rejected under 35 U.S.C. 103(a) as being unpatentable over Schuster et al (US Patent No. 6,483,600) in view of Blomfield-Brown (US Patent No. 5,625,678) and further in view of the admitted prior art disclosed in the specification on pages 1-4 and Fig.2.

With respect to claim 14, Schuster discloses (col. 9, line 36-53 and Fig. 2) internal architecture for the data network gateway 30 and 70 for use in a number of different types of applications such as Internet access, Internet telephony, facsimile transmissions, etc.... including telephone interfaces 34a-c, fax/voice modem 40a-c, and a data network interface 46 which contains software and hardware modules to perform call routing, modem configuration, and other features (a DSP module for carrying a user-initiated telephone conversation on a telephone line connecting the first telephone device and the second telephone device through the packet switching network).

Schuster discloses in Fig. 1a, a data network facsimile system for transmission digitized facsimile signals from facsimile device 20 to facsimile device 80 over the packet switching network (the DSP module responsive to analog fax signals and operative to convert analog fax

signals to digital fax signals and to packetize the digital fax signals for transmission through the packet switching network, to the second fax machine).

Schuster does not disclose fax transmission from the first fax machine to the second fax machine takes place on the telephone line causing a temporary interruption to the telephone conversation thereby avoiding the need for telephone connection to be disconnected prior to the fax transmission.

Blomfield-Brown discloses (col. 2, line 15-21) that when a person wants to send data (fax transmission) to the other person on a call, the sending modem temporarily mutes the handset and sends a signal directing the receiving modem to switch to data mode. When the receiving modem receives the signal, it mutes the handset and prepares to receive data. After transferring the data, both modems unmute their handsets and normal conversation ensues.

It would have been obvious to one having ordinary skill in the art at the time the invention was made to include the feature of temporary muting the telephone conversation, sending and receiving data while the conversation is on hold in Schuster's system, as suggested by Blomfield-Brown, to allow multiple applications such as voice and data running and sharing at the same time to increase the productivity and maximize the usage of such system.

Schuster does not disclose wherein frequency adjustments are made to compensate for differences in frequency between the fax transmission and the telephone signal. The admitted prior art discloses on page 4, lines 1-5 and Fig. 2 that in fax transmissions, a codec loaded into the DSP followed by an "overlay". The "overlay" converts the rate of transmissions of fax signals to the appropriate speed necessary for transmission of fax information over IP. It would have been obvious to one having ordinary skill in the art at the time the invention was made to

include the overlay feature in Schuster's system, as suggested by the admitted prior art, to compensate for frequency differences in a network.

With respect to claim 15, the limitation recited in claim 15 is addressed in the rejection of parent claim 14; Wherein, Blomfield-Brown discloses that when a person wants to send data to the other person on a call, the sending modem temporarily mutes the handset and sends a signal (a fax overlay is transferred between the router device and the remotely-located prior prior to transmission of fax information) directing the receiving modem to switch to data mode. When the receiving modem receives the signal, it mutes the handset and prepares to receive data. After transferring the data, both modems un-mute their handsets and normal conversation ensues.

With respect to claim 16, the limitation recited in claim 16 is addressed in the rejection of parent claim 14; Wherein, Blomfield-Brown discloses that when a person wants to send data to the other person on a call, the sending modem temporarily mutes the handset and sends a signal (the router detects a fax tone prior to transmission of the fax information) directing the receiving modem to switch to data mode. When the receiving modem receives the signal, it mutes the handset and prepares to receive data. After transferring the data, both modems un-mute their handsets and normal conversation ensues (upon completion of the fax transmission the router device resumes the telephone conversation).

With respect to claim 20, Schuster discloses (col. 5, line 6-35) H.225 protocol is used in communications (connections are established pursuant to H.225 protocol).

Conclusion

5. The prior art made of record and not relied upon is considered pertinent to applicant's disclosure.

Panburana et al (US Patent No. 6,633,582 B1) discloses symmetrical codec selection in an asymmetrical codec environment.

6. Any inquiry concerning this communication or earlier communications from the examiner should be directed to Anh-Vu H Ly whose telephone number is 703-306-5675. The examiner can normally be reached on Monday-Friday 7:00am - 4:00pm.

If attempts to reach the examiner by telephone are unsuccessful, the examiner's supervisor, Chi Pham can be reached on 703-305-4378. The fax phone number for the organization where this application or proceeding is assigned is 703-872-9306.

Information regarding the status of an application may be obtained from the Patent Application Information Retrieval (PAIR) system. Status information for published applications may be obtained from either Private PAIR or Public PAIR. Status information for unpublished applications is available through Private PAIR only. For more information about the PAIR system, see http://pair-direct.uspto.gov. Should you have questions on access to the Private PAIR system, contact the Electronic Business Center (EBC) at 866-217-9197 (toll-free).

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